CLAIMS

What is claimed is:

1. A microphone array processing system for performance enhancement in noisy environments, the system comprising:

a plurality of microphones positioned to detect speech from a single speech source and noise from multiple sources, and to generate corresponding microphone output signals, one of the microphones being designated a reference microphone and the others being designated data microphones;

a plurality of bandpass filters, one for each microphone, for eliminating from the microphone output signals a known spectral band containing noise;

a plurality of adaptive filters, one for each of the data microphones, for aligning each data microphone output signal with the output signal from the reference microphone; and

a signal summation circuit, for combining the filtered output signals from the microphones, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio.

2. A system as defined in claim 1, and further comprising speech detection circuitry, for enabling the plurality of adaptive filters only when speech is detected.

3. A system as defined in claim 1, and further comprising speech conditioning circuitry coupled to the signal summation circuit, to reduce reverberation effects in the output signal.

4. A system as defined in claim 3, wherein each of the adaptive filters includes:

means for filtering data microphone output signals by convolution with a vector of weight values;

means for comparing the filtered data microphone output signals from one of the data microphones with reference microphone output signals and deriving therefrom an error signal; and

means for adjusting the weight values convolved with the data microphone output signals to minimize the error signal.

5. A system as defined in claim 4, wherein each of the adaptive filters further includes fast Fourier transform means, to transform successive blocks of data microphone output signals to a frequency domain representation to facilitate filtering.

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6. A method for improving detection of speech signals in noisy environments, the method comprising:

positioning a plurality of microphones to detect speech from a single speech source and noise from multiple sources, one of the microphones being designated a reference microphone and the others being designated data microphones;

generating microphone output signals in the microphones;

filtering the microphone output signals in a plurality of bandpass filters, one for each microphone, to eliminate from the microphone output signals a known spectral band containing noise;

adaptively filtering the microphone output signals in a plurality of adaptive filters, one for each of the data microphones, and thereby aligning each data microphone output signal with the output signal from the reference microphone; and

combining the adaptively filtered output signals from the microphones in a signal summation circuit, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio.

7. A method as defined in claim 6, and further comprising the steps of:

detecting speech received by the microphones; and

enabling the step of adaptively filtering the microphone signals only when speech is detected.

8. A method as defined in claim 6, and further comprising the step of conditioning the combined signals in speech conditioning circuitry coupled to the signal summation circuit, to reduce reverberation effects in the output signal.

9. A method as defined in claim 8, wherein the step of adaptively filtering includes:

filtering data microphone output signals by convolution with a vector of weight values;

comparing the filtered data microphone output signals from one of the data microphones with reference microphone output signals and deriving therefrom an error signal;

adjusting the weight values convolved with the data microphone output signals to minimize the error signal; and

repeating the filtering, comparing and adjusting steps to converge on a set of weight values that results in minimization of noise effects.

10. A method as defined in claim 9, wherein the step of adaptively filtering further includes:

obtaining a block of data microphone signals;

transforming the block of data to a frequency domain using a fast Fourier transform;

filtering the block of data in the frequency domain using a current best estimate of weighting values;

comparing the filtered block of data with corresponding data derived from the reference microphone;

updating the filter weight values to minimize any difference detected in the comparing step;

transforming the filter weight values back to the time domain using an inverse fast Fourier transform;

zeroing out portions of the filter weight values that give rise to unwanted circular convolution; and

converting the filter values back to the frequency domain.

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